User Manual

Digital Processor

DMS48

APG

V 1.1

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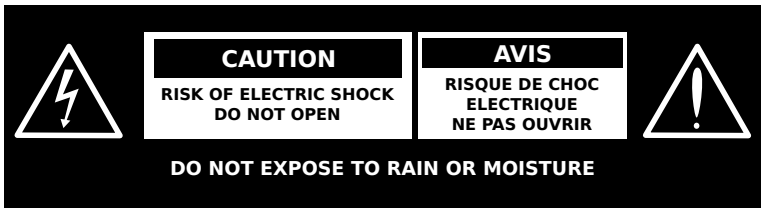
1 Important safety instructions



Caution: Risk of electric shock do not open !

Caution: In order to reduce the risk of electric shock, do not attempt to open any part of the unit. no user-serviceable parts inside. refer servicing to qualified service personnel.

Warning: To reduce the risk of fire or electric shock, do not expose this apparatus to rain or moisture. objects filled with liquids, such as vases, should not be placed on this apparatus to completely disconnect this apparatus from the ac mains, disconnect the power supply cord plug from the AC receptacle

The mains plug of the power supply cord must remain readily accessible

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water.
6. Clean only with dry cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Do not install near any heat sources such as radiators, heat registers, stoves or other apparatus (including amplifiers) that produce heat.
9. Do not defeat the safety purpose of the polarized or grounding type plug. A polarized plug has two blades with one wider than the other. A grounding type plug had two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.

10. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles and the point where they exit from the apparatus.
11. Only use attachments / accessories specified by the manufacturer.
12. Unplug this apparatus during lightning storms or when unused for long periods of time.
13. Refer all servicing to qualified service personnel. Service is required when the apparatus has been damaged in any way, such as power-supply cord or plug damaged, liquid has been spilled or objects have fallen into the apparatus, this apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.

Safety Warnings

- Permanent disconnection from the mains supply is to be achieved by removing the supplied cord connector from the back of the unit.
- Do not remove any covers, loosen any fixings or allow items to enter any aperture.
- Objects filled with liquids should not be placed on this apparatus.
- Replace the mains fuse only with a fuse of the same type

Installation instructions

1. **This product must be earthed**. Use only a flexible cable or cord provided with a green or green and yellow core which must be connected to the protective earthing terminal of a suitable mains plug or the earthing terminal of the installation. The cord must be of maximum length 7.5 meters, rated SJ, SJT, or SJE, 10A minimum and be marked VW-1.
2. Wiring to this product must only be made by suitably qualified personnel and must comply with all local requirements.
3. Do not install this equipment in an enclosed space. Do not limit free ventilation and movement of air around the back panel.
4. Only use attachments and accessories approved by or specified by the manufacturer.

For customers in Europe

This product complies with both the LVD (electrical safety) 73/23/EEC and EMC(electromagnetic compatibility) 89/336/EEC directives issues by the commission of the European community. Compliance with these directives implies conformity with the following European standards:

- EN60065 Product safety
- EN55103-1 EMC emissions
- EN55103-2 EMC immunity

This product is intended for the following electromagnetic environments: E1, E2; E3 & E4.

For customers in the USA

This product has been tested for electrical safety and complies with: UL60065 7$^{\text{th}}$ edition.

Federal Communications Commission Notice This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy, and if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try and correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the distance between the equipment and the receiver.
- Connect the equipment to an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

FCC Caution: Any changes or modifications not expressly approved by the party responsible for compliance could void the users authority to operate this equipment. This equipment has been designed to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment.

For customers in Canada

This product has been tested for electrical safety and complies with: CA /CSA C22.2 No.60065-03

Declaration of Conformity with Canadian ICES-003 This Class A digital apparatus complies with Canadian ICES-003.

2 Thanks and Unpacking

Thank you for choosing the APG DMS48 processor for your application. Please spare a little time to study the contents of this manual, so that you obtain the best possible performance from this unit. All APG products are carefully engineered for world-class performance and reliability. If you would like further information about this or any other APG product, please contact us. We look forward to helping you in the near future.

2.1 Unpacking the APG DMS48

After unpacking the unit please check carefully for damage. If damage is found, please notify the carrier concerned at once. You, the consignee, must instigate any claim. Please retain all packaging in case of future re-shipment.

3 Introduction and Key Features

3.1 Introduction

The APG DMS48 processor represents current state-of-the-art technology. Taking advantage of the latest advances in analogue to digital conversion and digital signal processing technologies the units achieve performance levels higher than previous devices. Below is a list of key features, followed by some information on the major advancements of the DMS48 feature set.

3.2 Key Features

- APG minimal signal path design
- Bandwidth: 96kHz sampling frequency provides for a nominally flat response beyond 40kHz.
- Three rotary encoders, illuminated buttons and graphical display provide a rapid, intuitive and user-friendly control interface
- High speed capable and flexible Ethernet communications that supports DHCP, static-IP and auto-IP and direct connection to a computer without the need for a router or a switch
- Class leading sonic performance achieved by the use of state of the art converters, a 4^{th} Generation Analogue Devices Sharc DSP and highly advanced DSP algorithms
- Powerful Drive Module concept, abstraction from device centric to speaker based control
- Unique VX limiter providing dynamic control for passive 2-way enclosures
- Unique LIR linear phase crossover shapes giving FIR-like performance without the drawbacks
- Linear phase HF system EQ profiling which provides perfect integration between enclosures
- Innovative excursion control limiter with sliding High Pass Filter; retains dynamic impact whilst effectively protecting drivers.
- Transducer thermal modelling provides regulation limiters, addressing long term overload
- Overshoot limiter governs amplitude of transient signals retaining average power whilst constraining peak energy
- Dante audio networking with AVB upgrade path
- AES3 inputs and outputs switchable in pairs
- High performance universal mains switch mode power supply, designed in-house.

3.3 Drive Modules

The DMS48 processor has a new way of ordering and grouping channels in order to give a more speaker based approach to controlling, designing and recalling speaker configurations; these are called Drive Modules. A Drive Module is the Processing provided by one Input DSP Block, and a number of Output DSP Blocks, which are associated with one-another by means of routing. For example, if Input DSP Block B is routed to Outputs 3 and 4, then this is a 2-way Drive Module with Input DSP Block B forming the Master control, and Output DSP Blocks 3 and 4 providing the driver-related control. Overall, this forms the control typically for one loudspeaker sub-system. The PWAPG Drive Module control panel for this sub-system may then be used for control and monitoring. The Presets in the DMS48 are Drive-Module centric, and are used to configure individual Drive Modules rather than the whole device. Importantly, Drive Modules move the focus away from the processing device, and onto the loudspeaker systems.

3.4 LIR Linear Phase Crossover Filtering

The DMS48 also includes a new type of crossover filtering Linea Impulse Response (LIR) crossover filtering, which gives a Linear Phase crossover that has a constant delay regardless of frequency (unlike other types of crossover which delay different frequencies to a different extent, thus smearing the arrival time). The LIR crossover can thus be described as having a flat Group Delay response, and thus entirely free of Group Delay Distortion. The shape of the LIR crossover filter is quite similar to a 4^{th} order or 24dB/Oct Linkwitz-Riley filter, and maintains zero phase difference between the adjacent bands across the crossover region to keep the polar response rock steady. See illustration 1.

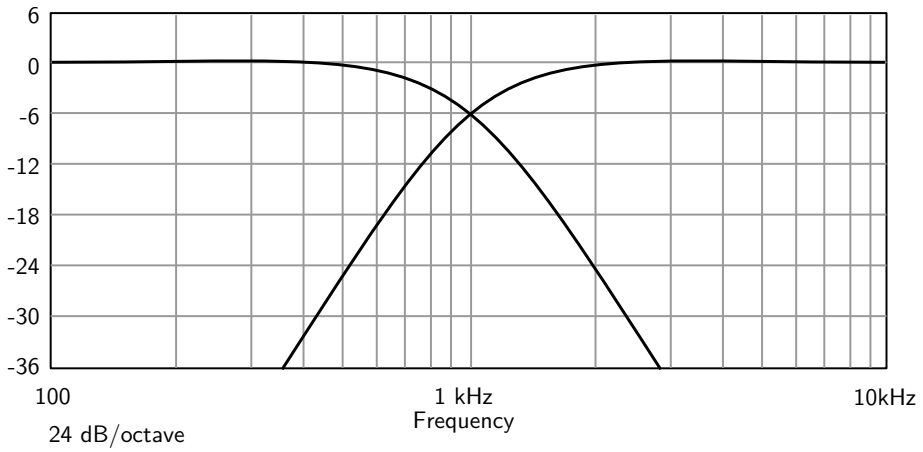


Figure 1: LIR Filter

3.5 FIR Linear Phase Equalisation

The Input High-Shelf Equalisers use Finite Impulse Response (FIR) filtering to produce Linear Phase equalisation; that is all frequencies are delayed by the same amount, perfectly preserving the transient response. This can also be important in applications where different amounts of EQ are applied to different parts of a speaker cluster, such as to add 'Throw' EQ boost so that parts of cluster which are throwing further can have HF absorption correction added. If

this EQ is not linear phase, then the zones where the speakers combine may suffer frequency response anomalies.

The User Guide

This user manual gives a progressively more detailed description of the functions of the APG DMS48 processor. A single page quick reference guide is provided for those users who are experienced with this type of equipment and just need to know how to drive the front panel. A detailed explanation of the front and rear panel controls and indicators is contained in the next section. The final section describes each individual function or feature with annotated images explaining its use. Where appropriate, the graphical display is shown to further elaborate on the units operation. To complete the manual a reference section is included, describing the technical performance of the device complete with graphs of filter responses and details of the Drive Module presets and their configuration.

A Menu Buttons

There are three buttons to determine which section of the device to view or edit. The <**OUT**> button displays pages of parameters associated with a particular output channel. The <**IN**> button displays pages of parameters associated with a particular input socket or input DSP channel. Pressing <**IN**> or <**OUT**> buttons repeatedly will scroll through the inputs/outputs of the processor. After the last channel, navigation returns to the default screen. The <**UTIL**> button displays pages of miscellaneous parameters not associated with any particular channel. Whilst in Edit mode, one of these three buttons will be illuminated. They are mutually exclusive pressing one of the buttons will deselect any others that are active.

A.1 Front Panel

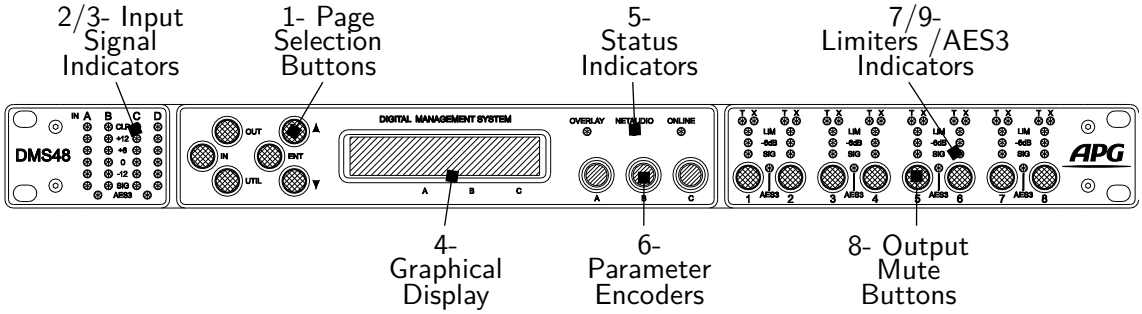


Figure 2: Front panel schema

A.1.1 Page Selection Buttons

When one of the buttons <**IN**>, <**OUT**> or <**UTIL**> is illuminated, the up <▲> and down <▼> arrows will also illuminate, informing the user that these buttons may be used to scroll through the various pages of parameters that may be viewed and edited. The <**ENT**> button is used to confirm an operation such as storing or recalling a preset or snapshot

A.1.2 Input Signal Indicators

A set of five indicators show **Sig**, **-12**, **0**, **+6** and **+12** for each of the DSP inputs **A B C D**. The signal present Indicators operate at approximately 40 dBu, giving a useful indication of

even relatively low input signal levels. The **0** indicators are intended to show nominal operating level and can also be useful for setting system gain structure.

A.1.3 Input Clip/mute Indicator

The input clip/mute indicator **CLP** warns the user of either input mute or input overload at +19 dBu. The overload indicator will only come on when all other signal lights are illuminated. If the other lights are off, this indicator shows a muted input state. Input mute is turned on/off in the <**IN**> gain page.

This differs from the **CLP** indication, as this remains illuminated at all times whilst mute is active.

A.1.4 Graphical Display

When the device is switched on, it will show the default screen. This provides a useful overview of channel allocation and Drive Module presets. The screen contrast can be changed by pressing the <**UTIL**> button to navigate to Screen and using encoder **A** will change the percentage; this can also help with the viewing angle. In most pages the currently selected channel and parameter information is displayed on the upper part of the screen and the parameter value on the lower part of the screen.

A.1.5 Status Indicators

The **OVERLAY** indicator shows when there are parameters active on a group layer, which the user cannot access through the front panel of the device. The **NETAUDIO** indicator shows that a networked digital audio card is installed and routed (Such as Dante or AVB). The **ONLINE** indicator has three states:

Off The unit is offline and not connected to a computer or network.

Flashing The unit is searching for an IP address, if the unit does not find an IP address the unit will assign itself an IP address automatically and the indicator will stop flashing.

On The unit is online and connected with software. IP settings can be viewed or changed within the <**UTIL**> pages.

A.1.6 Parameter Encoders

Three velocity sensitive parameter encoders are used to adjust parameters shown on the display. Up to three parameters at a time are displayed on the screen. The parameter name is shown above the parameter value in each of the three screen sections.

A.1.7 AES3 Active Indicators

The **AES3** active indicators show that the AES3 input/output is active for that pair of channels. AES3 transmits two channels down one XLR cable. On the DMS48 each pair is carried by the odd number connector such as, 1+2 will be carried on XLR connector 1 and 3+4 will be carried on XLR 3 and so on.

A.1.8 Mute Buttons

DSP output mute status is indicated and controlled by illuminated buttons for each channel.

A.1.9 Limiter Indicators

The limiter indicators are a collection of indicators which signal different information about the limiter status. The <**SIG**> indicator shows signal presence and will illuminate when a signal above - is present in the output. The second indicator <**-6dB**> shows that the signal has reached 6dB below the limiter threshold. The third <**LIM**> indicator signals that the threshold of that output channel has been reached. The <**X**> indicator also refers to the excursion limiter and will illuminate when the threshold of the excursion limiter has been reached. This <**X**> also indicates a signal 6dB higher than the limiter threshold. The <**T**> indicator shows that the power limiter is active, which will prevent long term thermal damages. Please note that because of the long release time of the thermal limiter, the <**T**> indicator may remain illuminated for several seconds after signal on that channel is reduced. The <**T**> indicator illuminates when the limiter is in at least 1dB of limit.

A.1.10 Selection Indicators

Each page that is visited when navigating will usually show either one, two or three parameters labelled **A**, **B** and **C**, each of which can be edited by turning the Encoder associated with that parameter also labelled **A**, **B** and **C**.

A.2 Rear panel

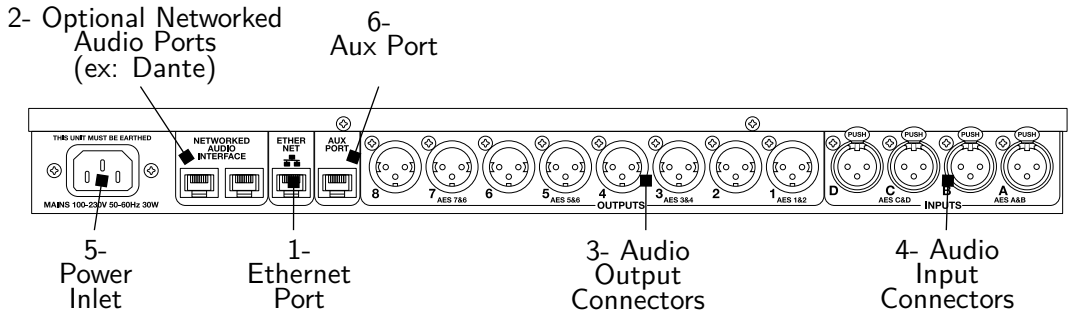


Figure 3: Rear panel

A.2.1 Ethernet Communications Port

APG DMS48 processor may be controlled entirely from another controller, typically a Personal Computer, running an application that is compliant with the ObCom standard such as PWAPG. Connection will normally be made to the controller via this Ethernet port connector. This port is also used for updating the firmware in the unit.

A.2.2 Networked Audio Ports

The DMS48 has the option for networked audio ports; if none are required a blanking plate will be fitted. There are several options for networked audio including Dante and AVB. For a full list please consult your vendor.

A.2.3 Audio Output Connectors

The processed outputs are impedance balanced, and wired; pin-1 directly bonded to ground as required by the AES48 standard, pin-2 hot and pin-3 cold. An unbalanced input may be driven by connecting pin 3 cold to the ground connection at the source end. When AES3 operation is selected on an output pair, the Odd channel connector is used for both channels (e.g. Input 1 is used for channels 1 and 2).

A.2.4 Audio Input Connectors

All audio connections are fully balanced and wired; pin-1 ground as required by the AES48 standard, pin-2 hot & pin-3 cold. If an unbalanced source is used, a connection should be made between the pin-3 cold signal and the ground connection at the source end. When AES3 operation is selected on an input pair, the Odd channel connector is used for both channels (e.g. Input 1 is used for channels 1 and 2).

A.2.5 Power Inlet

The APG DMS48 unit should be connected to a suitable mains electricity supply using an earthed IEC C14 connection power lead. The processor has a switch mode power supply that is capable of operating with a nominal mains voltage of 85V to 240V, 50/60Hz without re-configuration.

Note: The DMS48 must be earthed to a suitable power earth; failure to do so may affect performance and/or operation and will invalidate warranty and could be potentially hazardous.

A.2.6 Auxiliary Port

The auxiliary port is used for recalling the first two snapshots stored.

Also see AUX Port on page 25

B Operation

B.1 Starting up the unit

The unit will power up as soon as power is applied to the IEC power inlet; there is no power switch. When power is present the unit will go through its start-up cycle - first all the indicators will illuminate then go off, while the screen displays the boot loader information. Next the screen will display the application firmware information, and all the mute buttons will illuminate. The display will then show the default screen showing drive module configurations, the mutes will then return to the state they were in when the unit was powered off.

B.2 Drive Module Presets

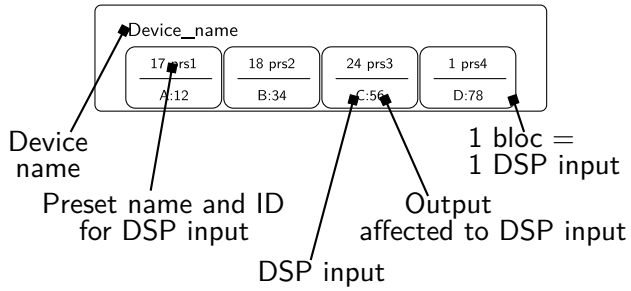


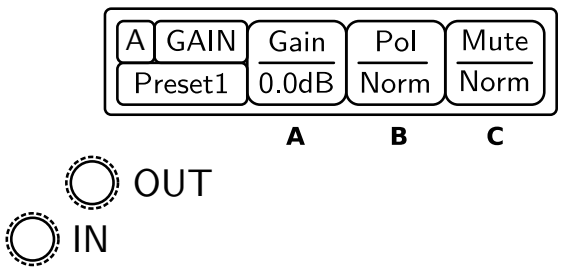
Figure 4: Main screen

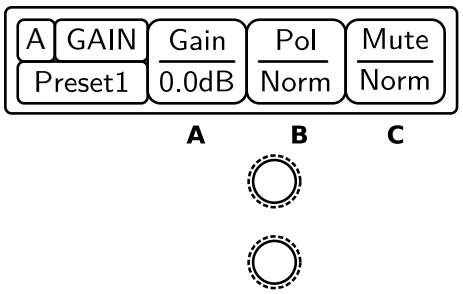
The DMS48 uses Drive Module presets, which are defined as a number of outputs driven from one DSP input. This system allows for better flexibility and greater functionality when loading and storing presets. Drive modules allow for a less processor centric and more speaker orientated system design. A drive module is the processing provided by one Input DSP, and a number of outputs, which are associated with one-another by means of routing. For example, if Input DSP B is routed to outputs 3 and 4, then this is a 2-way drive module with input DSP B forming the master control, and output DSP 3 and 4 providing the driver-related control. Overall, this forms the control typically for one loudspeaker sub-system. The Input DSP parameters then control the Drive Module, and the Drive Module control panel in the PWAPG application is then used for control and monitoring of this sub-system. Outputs assigned DSP Input letter Presets are Drive Module centric, in that a preset will change only the parameters in the given Drive Module (determined by which Input the preset is recalled to).

Note: DSP inputs are not the same as physical inputs. The DMS48 has four audio inputs and four DSP inputs. This is a matrix mixing system where any physical inputs, be they analogue, AES3 or networked audio feeds, can drive any number of DSP inputs.

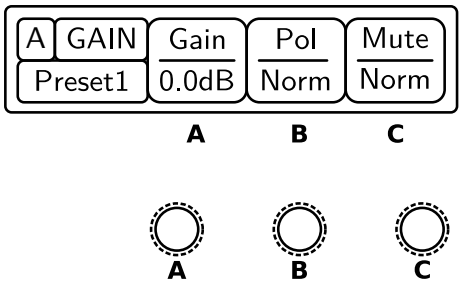
Also see Snapshots on page 24

B.3 Navigation and Parameters Edition





The DMS48 has 45 drive module preset locations and these can be stored and recalled from the <**IN**> pages, for the channel being viewed. To design a new crossover, press the desired <**IN**> or <**OUT**> button to enter the pages where the parameters for each of the channels are shown. Once in these pages the up <▲> and down <▼> buttons will scroll through the parameters for the inputs/outputs through the signal chain. Pressing the <**IN**> or <**OUT**> button repeatedly will scroll through the channels. This allows the user to enter all the parameters for each of the channels in the signal chain or enter each parameter for each channel one after another.



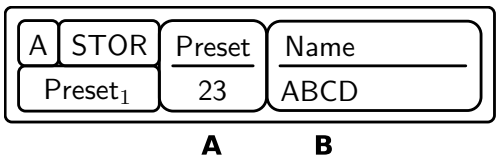
A parameter may be adjusted when it is displayed by turning one of the three-parameter encoders **A**, **B** or **C**. Each of the three-parameter encoders is associated with a zone on the display. Adjusting the leftmost parameter encoder **A** will change the value of the parameter showing in the leftmost zone of the display above the **A** and so on. Turning an encoder clockwise will increase the value of a parameter, or anticlockwise will decrease it. The encoders are velocity-sensitive so turning an encoder rapidly will cause the action to accelerate, so the value changes more rapidly.

B.4 Factory Presets

The device may contain a library of *Factory Presets* designed to suit a range of enclosures. *Factory Presets* contain some parameters that are fixed and hidden from view; the remainder of the parameters are available for user manipulation. The number and type of hidden parameters is dependent on the Factory Preset, typically crossover frequencies, output delay and some EQs are hidden; those settings that are a function of the loudspeaker cabinet design and should not require adjustment for different applications. *Factory Presets* are locked (as indicated by a box symbol after the Preset name) so they cannot be over-written. The user can, however, store an edited version of a Factory Preset in any free preset location. In addition to the *Factory Presets* the device may have further *Skeleton Presets* which will help to create new presets. They can be used to develop settings for any loudspeaker combination and are recalled in the

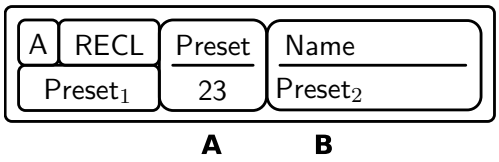
same way as the *Factory Presets* described above. These Presets are also usually locked but the user can name and store their own edited versions in any free preset location.

B.5 Storing Presets



Once a drive module has been created it can be stored by pressing the <**IN**> button until the edited channel is reached, then pressing the down <▼> button until **store** page is reached. Using encoder **A** will change the preset number. When the destination preset is reached, pressing the <**ENT**> button will enable the name associated with that preset to be changed. Once the name changing is active, the character to be changed will be highlighted and encoder **C** will edit the character. Using encoder **B** will move through the character positions. Once the new preset name has been entered, the operation can be confirmed by pressing the <**ENT**> button, then a message will be displayed, "ENT to confirm or ▼ to exit"; pressing <**ENT**> will store the preset.

B.6 Recalling Presets

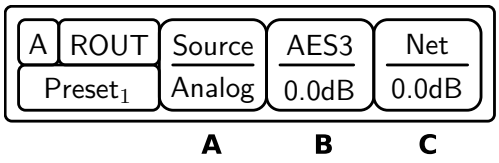


To recall a preset, press the <**IN**> button, then using the down <▼> button navigate to the RECL Preset page. Using the encoder, **A** will scroll through the presets available. When the desired preset is reached, pressing <**ENT**> will display the message "ENT to confirm or ▼ to exit", pressing <**ENT**> will recall the preset.

Also see Snapshots on page 24.

C Input

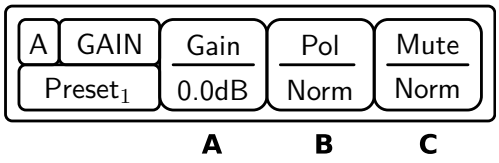
C.1 AES3 / Network Inputs



In addition to the usual analogue inputs, the DMS48 can also accept AES3 digital inputs. The same physical XLR sockets are used for both Analogue and AES3 inputs; the function of these being determined by the Source parameters in the Input Route menu. The Input sockets 1 and 3 are used to input pairs of AES3 channels, so adjusting the input to AES3 on either Input 1 or Input 2 will cause the other channel to change also. Similarly, adjusting the setting of Input 3 will affect Input 4 and vice-versa. When an input socket is selected

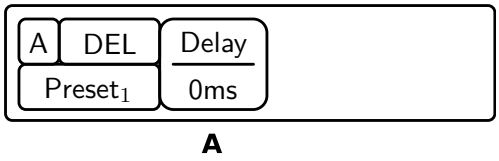
for AES3 operation, the AES3 indicator under the input meters will illuminate. So that the relative gain between an Analogue input and an AES3 input can be normalised, it is possible to adjust the gain of the AES3 channel using the Input Route AES3 Gain parameter. This device will automatically select the correct sample rate from the incoming stream between 28kHz and 108kHz. When the Digital Audio Network option (e.g. Dante) is fitted, it is also possible to select any channel(s) as being sourced from this network. To do this, connect the audio network connection to the connection on the rear of the DMS48, and set the relevant Input Route menu Source parameter to Dante (for example.) When the Digital Audio Network input is installed and routed, the Net Audio indicator above the encoders will illuminate. This indicator will come on even if there are no cables plugged into the networked audio port on the DMS48. Again, it is possible to set the relative gain between an Analogue input and the Digital Audio Network (Dante) using the Input Route Dante gain parameter. This device will automatically select the correct sample rate from the incoming stream. For other details on the operation of the Digital Audio Network, please refer to the relevant documentation.

C.2 Gain and Polarity



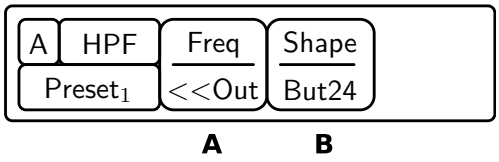
The gain page of the input channel selected allows users to increase or decrease the amount of signal going into the selected input. Using encoder **A** will change the value in 0.2dB steps from -40dB to +20dB. This page will also allow users to change the polarity of the selected input from normal to reverse, using encoder **B**. Using encoder **C** will mute the selected channel.

C.3 Delay



The delay page which controls the amount of delay associated with the input channel selected and is adjustable from 0 to 998ms. The delay parameter is adjustable in fine steps at low values; the adjustment becomes progressively coarser as the value increases.

C.4 High Pass Filter



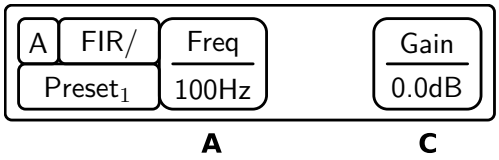
System high pass filtering is provided for the input signal. Filter type is selectable from 1^{st} order, Butterworth, Bessel, Linkwitz-Riley and Hardman. Filter slopes of up to 4^{th} order or 24dB / octave are provided. Not all filter types are available in all slopes. For example 18dB / octave Linkwitz-Riley filters do not exist. The Hardman type filter is always described by

its order as the filter becomes progressively steeper rather than following a linear slope so a dB/octave description is not accurate.

C.5 Parametric Equalisation

There are eight stages of equalisations available for each input channel, two shelving filters and six parametric filters.

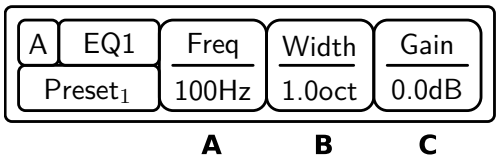
C.6 FIR Shelving EQ



The Input High Shelf EQ is implemented using a Finite Impulse Response (FIR) filter, and exhibits a linear phase response; that is all frequencies are delayed by the same amount. This can be important in applications where different amounts of EQ are applied to different parts of a speaker cluster, such as to add 'Throw' EQ boost so that parts of cluster which are throwing further can have HF absorption correction added. If this EQ is not linear phase, then the zones where the speakers combine may suffer frequency response anomalies. Being a linear phase FIR equaliser, this necessarily introduces some latency delay, which is constant regardless of the settings. However, when the frequency is set to maximum (Off), it is removed from the signal path entirely, so it does not add any latency. Using encoder **A** will change the frequency parameter from 2kHz to 20kHz and using encoder **C** will change the cut or boost in 0.2dB increments. The filter (and its associated latency) can be completely removed by setting the frequency parameter to the maximum off position.

Also see Latency Delays on page 25.

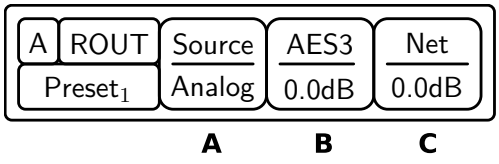
C.7 Parametric Filters



Parametric filters are defined by frequency, bandwidth and gain. The frequency is controlled by encoder **A** and ranges from 10Hz to 25.6kHz. The bandwidth, shown as Width on the screen, is controlled by encoder **B** and ranges from 0.10 octaves to 5.2 octaves. Bandwidth can be shown and adjusted as Q or Octaves (Oct). Gain is controlled by encoder C and is available in 0.2dB increments.

Also see Bandwidth Units on page 23.

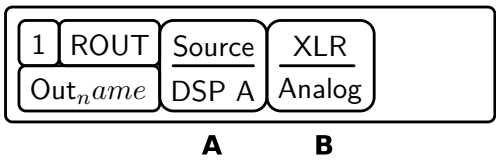
C.8 Routing



Routing allows users to route any physical or digital signal to any DSP input. This is a matrix mixing system where all DSPs can be driven from any one input or pairs of inputs **1+2** or **3+4**. Summed inputs have 6dB of attenuation so that a sum of largely similar programme material remains at the correct calibrated level. When a sum input is selected, the input DSP meters show the higher of the two inputs so that indication of the onset of clipping of either input is indicated.

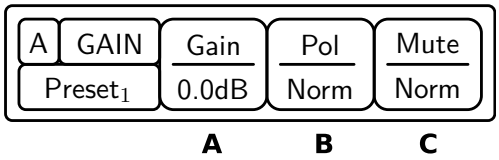
D Output

D.1 AES3 outputs



In addition to the usual analogue outputs, the DMS48 also allows outputs to be sent digitally using AES3. The same physical XLR sockets are used for both Analogue and AES3 outputs; the function of these being determined by the XLR parameters in the Output Route menu. The Odd numbered output sockets are used to output pairs of AES3 channels. When an output socket is selected for AES3 operation, the AES3 indicator under the output bar graphs will illuminate. The output sample rate for AES3 will always be 96kHz.

D.2 Gain and Polarity



The gain page of the output channel allows users to increase or decrease the relative signal gain for the selected output. Using encoder **A** will change the value in 0.2dB steps from -40dB to +20dB. This page will also allow users to change the polarity of the selected output from normal to reverse, using encoder **B**.

D.3 Delay

The delay page controls the amount of delay associated with the output channel selected and is adjustable from 0 to 998ms. The delay parameter is adjustable in fine steps at low values; the adjustment becomes progressively coarser as the value increases.

D.4 High and Low pass Filters

High pass crossover filtering is provided for the output signal. Filter type is selectable from 1^{st} order, Butterworth, Bessel, Linkwitz-Riley, Hardman and LIR Linear Phase using encoder B. Filter slopes of up to 8^{th} order or 48dB / octave are provided. Not all filter types are available in all slopes. For example 18dB / octave Linkwitz-Riley filters cannot be selected because they do not exist. The Hardman type filter is always described by its order as the filter becomes progressively steeper rather than following a linear slope so a dB/octave description is not accurate.

D.5 LIR Crossover Filtering

Unique to APG, Linea Impulse Response (LIR) crossover filtering gives a Linear Phase crossover which has a constant delay regardless of frequency (unlike other types of crossover which delay different frequencies to a different extent, thus smearing the arrival time). The LIR crossover can thus be described as having a flat Group Delay response, and thus entirely free of Group Delay Distortion, this is exactly the same as can be provided by common FIR filtering but without the complications and disadvantages inherent with the FIR technique. The shape of the LIR crossover filter is similar to a 4^{th} order Linkwitz-Riley filter, and maintains zero phase difference between the adjacent bands across the crossover region to keep the polar response rock steady. Linear Phase filtering necessarily introduces delay; the laws of physics demand it. To keep this delay to a minimum, it is recommended that more conventional crossover shapes (such as Linkwitz-Riley) are used for the very lowest frequency high-pass edge, particularly if this is less than perhaps 100Hz, which is well below the frequency thought to cause audible Group Delay Distortion. This constant delay will depend on the lowest high-pass frequency used in the crossover filters in a given Drive Module.

Also see Latency Delays on page 25.

D.6 Parametric Equalisation

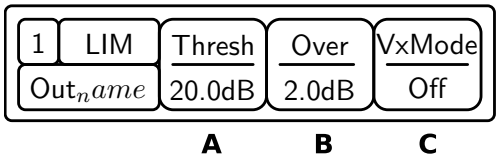
There are ten different EQ filters, two shelving filters and eight parametric filters. Parametric filters are defined by frequency, bandwidth and gain. The frequency is controlled by encoder **A** and ranges from 10Hz to 25.6kHz. The bandwidth, shown as Width on the screen, is controlled by encoder **B** and ranges from 0.10 octaves to 5.2 octaves. Bandwidth can be shown and asjusted as Q or Octaves (Oct). Gain is controlled by encoder **C** and is available in 0.2dB increments.

Also see Bandwidth Units on page 23.

D.7 Limiters

The DMS48 includes three limiters in the output signal path.

D.7.1 Vx Limiter



This is a peak-detecting signal limiter. The VX Mode parameter determines the style of limiter. When Virtual Crossover (VX) mode is off, the limiter is controlled in a conventional manner, the only controls being Threshold and Overshoot. The Overshoot limiter prevents the signal from exceeding threshold during the attack phase of the main limiter by more than a predetermined amount. The optimal Overshoot setting is usually about 8dB. Lower Overshoot settings will sound progressively harder. When VX mode is engaged, the user can choose the crossover point of a virtual crossover, which gives two limiters per output so the user can individually limit the drivers in a passive 2-way enclosure using individual thresholds, and optimised attack and release characteristics for each. The Threshold of the second Hi limiter, is set relative to the threshold of the first Lo limiter. The effect of the VX threshold and split frequency on the Limiter curve can be seen in PWAPG. This Limiter introduces some delay. In

non-VX mode, this delay will depend on the lowest high-pass frequency used in the crossover filters in a given Drive Module. In VX mode, the delay is related to the Split frequency. This delay will be applied to all of the outputs in a given Drive Module to keep them in phase.

Also see Latency Delays on page 25.

D.7.2 Tmax Thermal Limiter

The Thermal Limiter is intended to protect the driver against damage due to over-heating. This protection is calibrated by APG and cannot be modified. *Also see Amplifier Gain on this page.*

D.7.3 Xmax Excursion Limiter

The Excursion Limiter protects the driver against excessive linear movement of the cone and voice-coil which could otherwise cause mechanical damage. This protection is calibrated by APG and cannot be modified. *Also see Amplifier Gain on the current page.*

D.7.4 Amplifier Gain

Expressed in dB, this is the gain of the amplifier which the output of the DMS48 is feeding. This value must be entered to allow the DMS48 to correctly calibrate the thermal and excursion limiter for the amplifier in use.

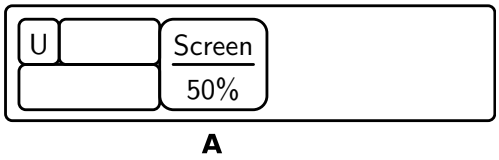
D.8 Routing

Outputs can be driven from any DSP input. This routing is the fundamental means by which Drive Modules are created.

Also see Drive Modules on page 9.

E Utility Pages

E.1 Screen Contrast

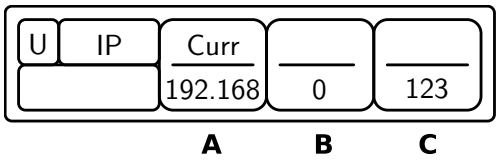


The Screen page in the **UTIL** section adjusts the contrast (and optimal viewing angle) of the screen from 0-100% in 1% increments using encoder **A**.

E.2 Stereo Linking

Stereo linking is available between DSP Drive Modules A + B and C + D. Changing a parameter in either of the stereo linked Drive Modules will change the other linked Drive Module. Stereo linking is controlled by the **STEREO** page in the **UTIL** menu.

E.3 Current Ethernet IP Address



This may be viewed in The **IP Curr** page of the **UTIL** menu. This value cannot be edited.

E.4 IP Mode

The Ethernet IP address may be automatic **Auto** or may be a fixed static value **Static** as determined by the IP Mode page of the **UTIL** menu.

Warning Do NOT use Static mode unless your IT system specifically requires it. Auto mode should always be used where possible since in this mode, the DMS48 device can always be discovered by the PWAPG application. When in Static mode, a warning will flash on the Default screen.

Also see Ethernet on the next page.

E.5 IP Static

This allows the Static Ethernet IP address to be adjusted by the three rotary encoders the the IP Static page of the **UTIL** menu. This will only have effect when in Static mode.

E.6 Store Snapshot

This page of the **UTIL** menu allows a Snapshot to be defined.

Also see Snapshots on the following page.

E.7 Recall Snapshot

This page of the **UTIL** menu allows a Snapshot to be recalled.

Also see Snapshots on the next page.

E.8 Bandwidth Units

This page of the **UTIL** menu allows the Bandwidth of Parametric Equalisers to be viewed and adjusted in either Octaves or Q.

E.9 Latency display

This page displays the latency induced by signal processing in the device. This latency value is computed, and not directly measured. It also does not take into account the additionnal delays, added either by the user or by APG factory presets.

Also see Latency Delays on page 25.

F Ethernet

F.1 Ethernet configurations

IP addressing in the Device can be completely automatic; No setup is required. When first installing and launching PWAPG, the computer Firewall may ask to allow PWAPG to access the network.

Note: This must be allowed.

F.2 DHCP

There are two primary IP address ranges one used when there is a DHCP server present (e.g. a Router in the system), and another where there is no DHCP server (so the Device and the Computer will instead use 'Auto IP'). Both the device and the computer must be in the same IP address range. In an office IT environment where there is a DHCP server, both the computer and the Device will be in the DHCP IP address range, so will connect immediately.

F.3 AUTO-IP

After the computer is switched on in an isolated network, or unpluged from an IT network, it takes a while to time out of DHCP searching, so will not connect immediately. It can take several minutes for the computer to acquire an Auto IP address. Similarly, the Device will initially search for a DHCP server when first switched on (during which time its Online Indicator will be flashing). It can take one minute before it will be able to connect while is acquires an Auto IP address.

F.4 Static-IP

If the device or the computer has a static IP address set, PWAPG may not be able to 'see' the device if it is in a different IP Address range - unless there are good reasons it is best to avoid the use of static IP addressing if at all possible.

F.5 IP Troubleshooting

If PWAPG cannot connect to the device:

- Check that the Firewall in the computer will allow PWAPG access to the network.
- Wait 10 minutes (for the computer to acquire the correct IP address) and try again.
- Check that the Current IP address in the device is compatible with the IP address of the computer. Generally, the leftmost two sets of 3 digits should be the same.

G Snapshots

The snapshot menu is accessed by the UTIL pages and allows users to trigger a set of drive modules and other device-wide settings to effectively recall a processor-wide preset. Snapshots stored from the device will include all of the Drive Modules and other settings. If Snapshots are created in PWAPG, users can define how much information and which settings they wish to store in the snapshot. In addition to being able to recall snapshots from PWAPG and

the DMS48's front panel, the first two snapshots are also recallable from the auxiliary port using a binary combination of pins. The parameters inside Drive Module Presets are not individually stored in Snapshots, so recalling a Snapshot will merely trigger the recall of the appropriate Drive Module presets, rather than restoring the parameters that were active when the Snapshot was stored. This has the distinct advantage that the library of OEM presets may be updated without having to be concerned about what parameters might have been saved in users Snapshots. It does however require that any existing edits to Drive Modules are stored before a Snapshot is stored.

G.1 AUX Port

Either type of Snapshot (Partial or Full) may be recalled either via the device User Interface, from the PWAPG application or via the AUX port on the rear of the DMS48. Snapshot 1 or Snapshot 2 may be recalled using a binary combination of selections on the AUX port of the device as follows:

AUX X	AUX Y	ACTION
Open	Open	No Change
Gnd	Open	Recall Snapshot 1
Open	Gnd	Recall Snapshot 2
Gnd	Gnd	Recall Snapshot 2

The above allows the AUX Port to be used either in Event mode (whereby a momentary push-button or momentary relay contact closure on one of the AUX lines will cause a Snapshot to be recalled). Or in State mode (where a rotary switch etc. connected to both of the AUX lines selects which Snapshot to use).

Note: If the contact closure port is in use, this will override the setting in the menu.
Also see Snapshots on the preceding page.

H Latency Delays

All Digital Signal Processing, and conversion between different formats of signal analogue/ digital/ network etc., necessarily introduces some delay (latency) to the signal path. Of course, we strive to minimise these latencies. Small as they are, it is sometimes useful to know their precise values. The following describes the principal latencies introduced by various parts of the processing:

Analogue Input	0.385ms
OR Digital input at 96KHz sample rate	0.5ms
OR Digital input at 48KHz sample rate	0.66ms
Analogue Output	0.402ms
OR AES3 Output	0.1ms
OR Dante Output	0.5ms (TBC)
Input HiShelf FIR	0.4ms (0ms if set to Off)
LIR Linear Phase crossover	1.19ms/Fhp(kHz) limited to 30ms maximum (1)
VxLim Limiter (VX mode off)	0.12ms/Fhp(kHz) limited to 2ms maximum (1)
OR VxLim Limiter	(VX mode on) 0.358ms/Fsplit(kHz) limited to 9ms maximum (1)

(1) This latency figure is calculated using the Vx Split frequency (in kHz).

Example:

Analogue Input	0.385ms
Analogue Output	0.402ms
Input HiShelf FIR (On)	0.4ms
LIR Linear Phase crossover (Off)	0ms
VxLim Limiter (VX mode on, 1KHzFsplit)	0.358ms
Total	1.545ms

Please note that the latencies within a Drive Module are equalised among outputs of that Drive Module. That is, padding delay will be automatically added to some outputs such that the total latency is the same in each output of a Drive Module. This latency equalisation does not extend outside a Drive Module, so Drive Modules are not guaranteed to have the same latency as one-another.

The computed latency value is displayed in UTILS pages, see Latency display on page 23

H.1 Audio Network outputs

When the Digital Audio Network option (e.g. Dante) is fitted, the eight audio outputs are supplied to eight channels on the Audio Network, regardless of the setting of the Analogue/AES3 XLR selection. The sample rate will be determined by the sample rate detected on the network. For other details on the operation of the Digital Audio Network (such as Dante), please refer to the relevant documentation.

H.2 Secure Mode

When activated, this will disable all the front panel controls so they cannot affect the signal path, making the unit secure against tampering. When in secure mode, the indicators still operate normally. To activate or deactivate secure mode, push and hold one of the menu buttons. When in secure mode, this will be shown on the display. Note that the communications port is still active in secure mode.

I Technical Specifications

- Input impedance >10kΩ balanced
- Output Imp <100Ω imp. balanced
- Max Input level +20dBu
- Max Output level +18dBu into 600Ω
- Sample rate 96kHz
- AES3 Input sample rate 28kHz 108kHz
- AES3 Output sample rate 96kHz
- Frequency Response 10Hz - 40kHz
- Input Dynamic range >120dBa Typ.
- Output Dynamic range >118dBa Typ.
- THD (20Hz-20kHz) <0.008% Typ.
- Mains required 85-230VAC 50-60Hz
- Mains power 30W

I.1 Connectors

- Audio input 3 pin female XLR
- Audio output 3 pin male XLR
- Ethernet Shielded RJ45
- Aux Contact Phoenix Connector
- Mains 3 pin IEC

I.2 Environmental

- Temperature 0 to +45C
- Humidity 0 to 80% RH (non-condensing)

I.3 Dimensions

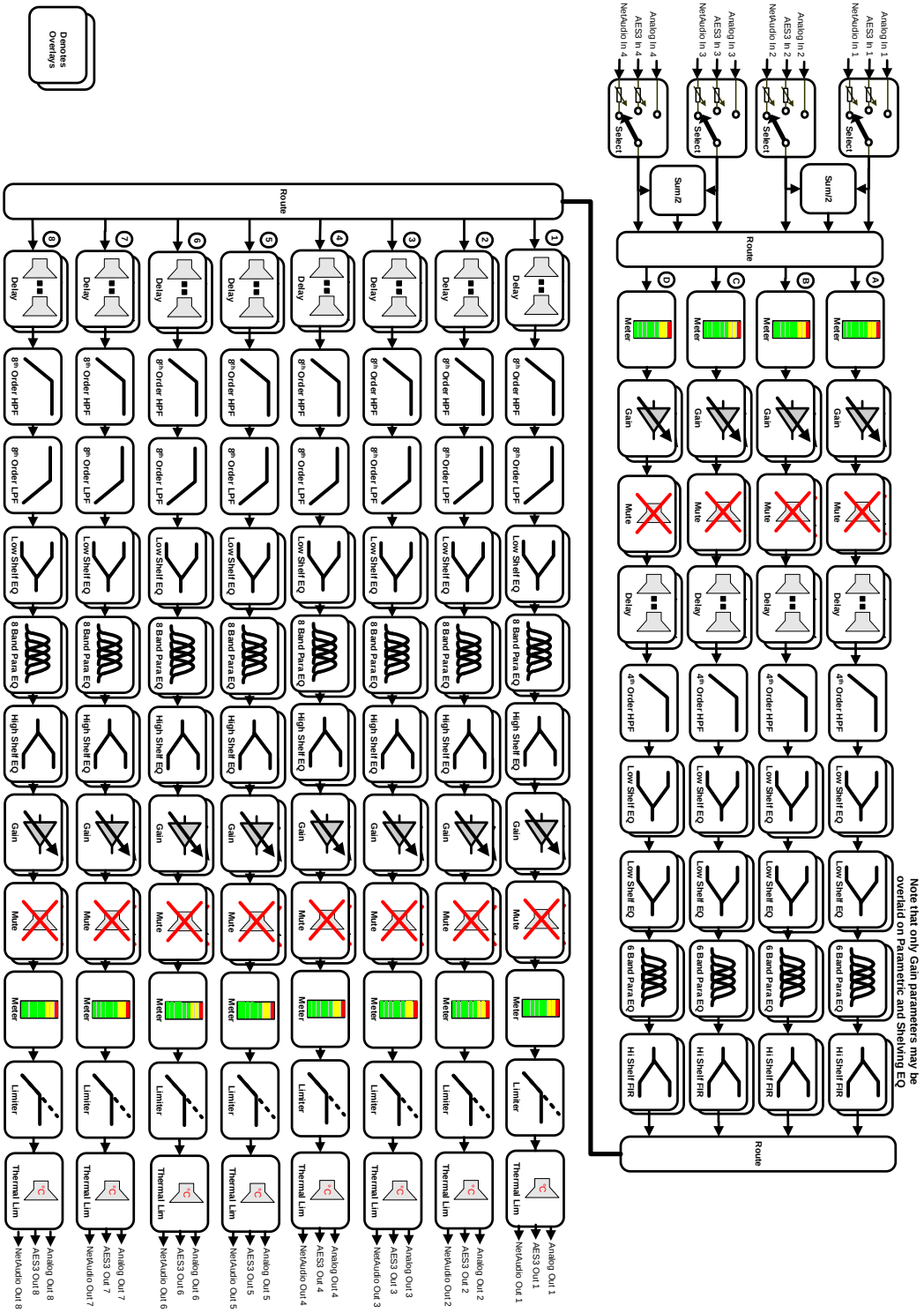
- Height 1U (44mm)
- Width 482mm
- Depth 254mm
- Weight 2.7kg net

I.4 Options

There is internal provision for digital audio network option cards to be fitted. Currently APG plan to support:

- DANTE
- AVB

I.5 DMS48 Synoptic view



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